REMARKS

Reconsideration and allowance of the application are respectfully requested. Claims 1-18 are pending and stand rejected. Applicant herein amends claims 1, 3, 5, 7, 10, 12 and 16. Claims 9, 11, 13-15, 17 and 18 are canceled. New claims 19-49 are added to more fully claim Applicant's invention. The specification is amended. No new matter has been introduced.

Amendments to the Specification

Applicant amends the specification to correct several typographic errors. Formula (1 in the specification as filed omitted an *n*/2 term. However, the specification clearly identifies formula (1 as the formula for "IMDCT," which is previously defined by the specification as the inverse modified discrete cosine transform. Moreover, the specification further states that "[f]or purpose of illustration, the operation of the audio decoder system 10 is described using MP3-encoded audio data ..." Specification as filed, page 7, lines 14-15. At page 2, lines 22 and 23, the specification refers to "[p]erceptual audio codecs, such as MPEG-1 Layer III Audio Coding (MP3), as specified in the International Standard ISO/IEC 11172-3 ..."

Under MPEP § 2163.07, section II., obvious errors in a specification may be corrected without introduction of new matter where a person skilled in the art would recognize the existence of error and the appropriate correction. Because the specification clearly indicated that formula (1 was intended to be the inverse modified discrete cosine transform, because the specification uses formula (1 as part of describing an error concealment process using MP3-encoded data, and because persons skilled in the art would understand that the IMDCT should be as set forth in the appropriate ISO/IED 11172-3 standard and not as set forth in the originally-

filed specification, no new matter has been introduced. A copy of the relevant page of ISO/IED 11172-3 setting forth the IMDCT formula is attached hereto. ISO/IED 11172-3 was also previously submitted in an Information Disclosure Statement.

The remaining specification amendments correct an obvious punctuation error and an incorrect reference number. The corrected reference number corresponds to a reference number in FIG. 5.

<u>The Claims</u>

The Office Action rejected independent claims 1 and 9 and dependent claims 2, 3, 6, 10, 11 and 13-16 as obvious based on U.S. Patent 5,852,805 (Hiratsuka et al.) in combination with U.S. Patent 5,256,832 (Miyake). Even if these references could properly be combined, which Applicant does not concede, they still fail to teach a recited feature of claim 1. Claim 1 as amended recites replacing at least a first part of an erroneous audio segment with a corresponding part of a stored audio bit stream portion, wherein the corresponding part is selected based on a time relationship between the first part and one of the $(k+1)^{th}$ and $(2k+1)^{th}$ beats. In other words, and as described in the specification, the first part of the erroneous segment has a particular time relationship to a beat defining the second interval. The part of the signal used to replace the first part of the erroneous segment is selected based on that time relationship. In this manner, and as described in the specification, the error correction is more representative of the original waveform than is the case using conventional methods.

Hiratsuka does not teach replacing an erroneous part with a <u>corresponding</u> part from another inter-beat interval. Instead, Hiratsuka teaches muting the output when an erroneous

segment occurs, replacement with a preceding portion that is not selected based on the relationship between an erroneous part and a beat, or otherwise concealing an error in a manner different from that of claim 1. Miyake similarly fails to teach replacing an erroneous part with a corresponding part from another inter-beat interval. Claims 2-8, 10, 12, 16 and 19-28 depend from claim 1, and are therefore allowable for at least the same reasons as claim 1.

Independent claim 29 is directed to a terminal, and also recites replacing at least a first part of an erroneous audio segment with a corresponding part of a stored audio bit stream portion, wherein the corresponding part is selected based on a time relationship between the first part and one of the $(k+1)^{th}$ and $(2k+1)^{th}$ beats. Accordingly, claim 29 (and its dependent claims 30-49) are allowable for at least the same reasons as claim 1.

The remaining rejections in the Office Action are moot in view of the above.

Finally, Applicant notes that "packet" was changed to "frame" in the claims for consistency with the specification, and that no change in claim scope is intended by this change.

It is respectfully submitted that this application is now in condition for allowance. Should the Examiner believe that anything further is desirable in order to place the application in even better form for allowance, he is respectfully urged to contact Applicant's undersigned representative at the below-listed number.

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INTERNATIONAL STANDARD ISO/IEC 11172-3:1993
TECHNICAL CORRIGENDUM 1

Published 1996-04-15

Information technology — Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s —

Part 3:
Audio

TECHNICAL CORRIGENDUM 1

Technologies de l'information — Codage de l'image animée et du son associé pour les supports de stockage numérique jusqu'à environ 1,5 Mbit/s —

Partie 3: Audio

RECTIFICATIF TECHNIQUE 1

Technical corrigendum 1 to International Standard ISO/IEC 11172-3:1993 was prepared by Joint Technical Committee ISO/IEC JTC 1, *Information technology.*

UDC 681.3.04(084.14) Ref. No. ISO/IEC 11172-3:1993/Cor.1:1996(E)

Descriptors: data processing, moving pictures, audio data, video recording, data storage, digital storage, coded representation, coding (data conversion), digital encoders.

channel. Above this bound decoding of intensity stereo is applied using the scalefactors of the right channel as intensity stereo positions. An intensity stereo position of 7 in one scalefactor band indicates that this scalefactor band is not decoded as intensity stereo.

```
Scalefactor bands :
|    |    |    |    |    |    |    |    |    |    | . . .    |    |
|<--- nonzero_part of spectrum (right chan) --->|<----- zero_part of spectrum ------>|
|<------- m/s or l/r stereo coded part ---------->|<- intensity stereo coded part ->|
```

For each scalefactor band (sb) coded in intensity stereo, the following steps are executed:

1) the intensity stereo position is_pos_{sb} is read from the scalefactor of the right channel.

2) if (is_pos_{sb} == 7) do not perform the following steps (illegal is_pos).

3) $is\_ratio = \tan\left(is\_pos_{sb} * \frac{\pi}{12}\right)$.

4) $L_i := L_i * \dfrac{is\_ratio}{1 + is\_ratio}$ for all indices i within the actual scalefactor band sb.

5) $R_i := L_i * \dfrac{1}{1 + is\_ratio}$ for all indices i within the actual scalefactor band sb.

2.4.3.4.10 Synthesis filterbank

Figure A.4. shows a block diagram including the synthesis filterbank. The frequency lines are preprocessed by the "alias reduction" scheme (see the block diagrams in in figure A.5 and in table B.9. for the coefficients) and fed into the IMDCT matrix, each 18 into one transform block. The first half of the output values are added to the stored overlap values from the last block. These values are new output values and are input values for the polyphase filterbank. The second half of the output values is stored for overlap with the next data granule. For every second subband of the polyphase filterbank every second input value is multiplied by -1 to correct for the frequency inversion of the polyphase filterbank.

2.4.3.4.10.1 Alias reduction

For long block_type granules (block_type != 2) the input to the synthesis filterbank is processed for alias reduction before processing by the IMDCT. The following pseudo code describes the alias reduction computation:

```
for (sb=1; sb<32; sb++)
    for (i=0; i<8; i++) {
        xar[18*sb-1-i] = xr[18*sb-1-i]Cs[i] -  xr[18*sb+i]Ca[i]
        xar[18*sb+i] = xr[18*sb+i]Cs[i] + xr[18*sb-1-i]Ca[i]
    }
```

The indices of arrays xar[] and xr[] label the frequency lines in a granule, arranged in order of lowest frequency to highest frequency, with zero being the index of the lowest frequency line, and 575 being the index of the highest. The coefficients: Cs[i] and Ca[i] can be found in table B.9. Figures A.5 and A.6 illustrate the alias reduction computation.

Alias reduction is not applied for granules with block_type == 2 (short block).

2.4.3.4.10.2 IMDCT

In the following, n is the number of windowed samples (for short blocks n is 12, for long blocks n is 36). In the case of a block of type "short", each of the three short blocks is transformed separately. n/2 values X_k are transformed to n values x_i. The analytical expression of the IMDCT is:

$$x_i = \sum_{k=0}^{\frac{n}{2}-1} X_k \cos\left(\frac{\pi}{2n}\left(2i + 1 + \frac{n}{2}\right)(2k+1)\right) \quad \text{for } i = 0 \text{ to } n-1$$